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Chan et al.

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(45) **Date of Patent: *Mar. 23, 2004**

(54) **PACKET NETWORK TELEPHONE INTERFACE SYSTEM FOR POTS**

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(*) Notice: This patent issued on a continued prosecution application filed under 37 CFR 1.53(d), and is subject to the twenty year patent term provisions of 35 U.S.C. 154(a)(2).

Subject to any disclaimer, the term of this patent is extended or adjusted under 35 U.S.C. 154(b) by 0 days.

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(52) **U.S. Cl.** **370/354**; 370/401; 379/88.17; 379/93.07; 379/900

(58) **Field of Search** 370/353, 354, 370/355, 356, 401, 466, 419; 379/88.17, 93.07, 900, 902

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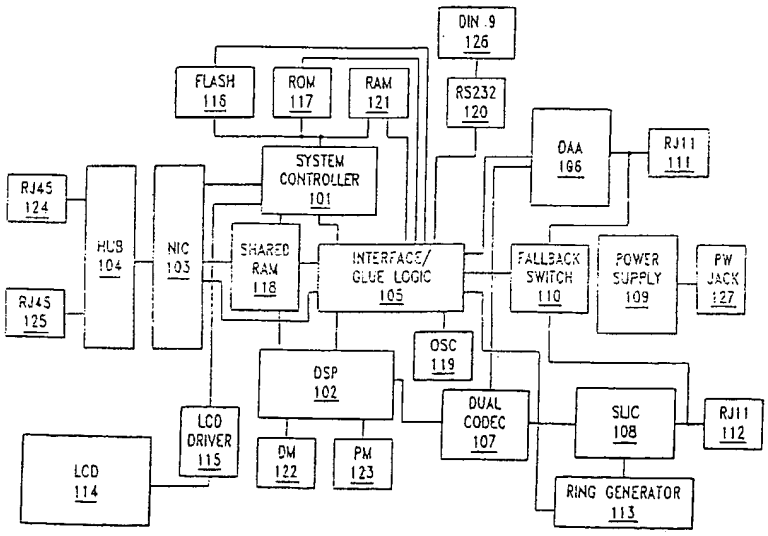
Assistant Examiner—Ahmad Elallam
(74) *Attorney, Agent, or Firm*—Scully, Scott, Murphy & Presser; Douglas W. Cameron

(57) **ABSTRACT**

The packet network phone unit is an interface between a telephone instrument and a packet network for enabling voice data to be communicated over the packet network. The unit also functions as a gateway between a packet network and a public switched telephone network, thereby interconnecting the networks and providing a public switched telephone network gateway to a packet network. When power is not supplied to the unit, a fallback switch automatically links the telephone instrument directly to the public switched network, bypassing the circuitry in the unit. The unit includes links to a plurality of external serial communications lines as well as an interface for connecting to a personal computer. The unit also includes an LCD driver and a display for displaying information such as a caller identification. The packet network phone unit also may function an answering machine/messaging system.

20 Claims, 7 Drawing Sheets



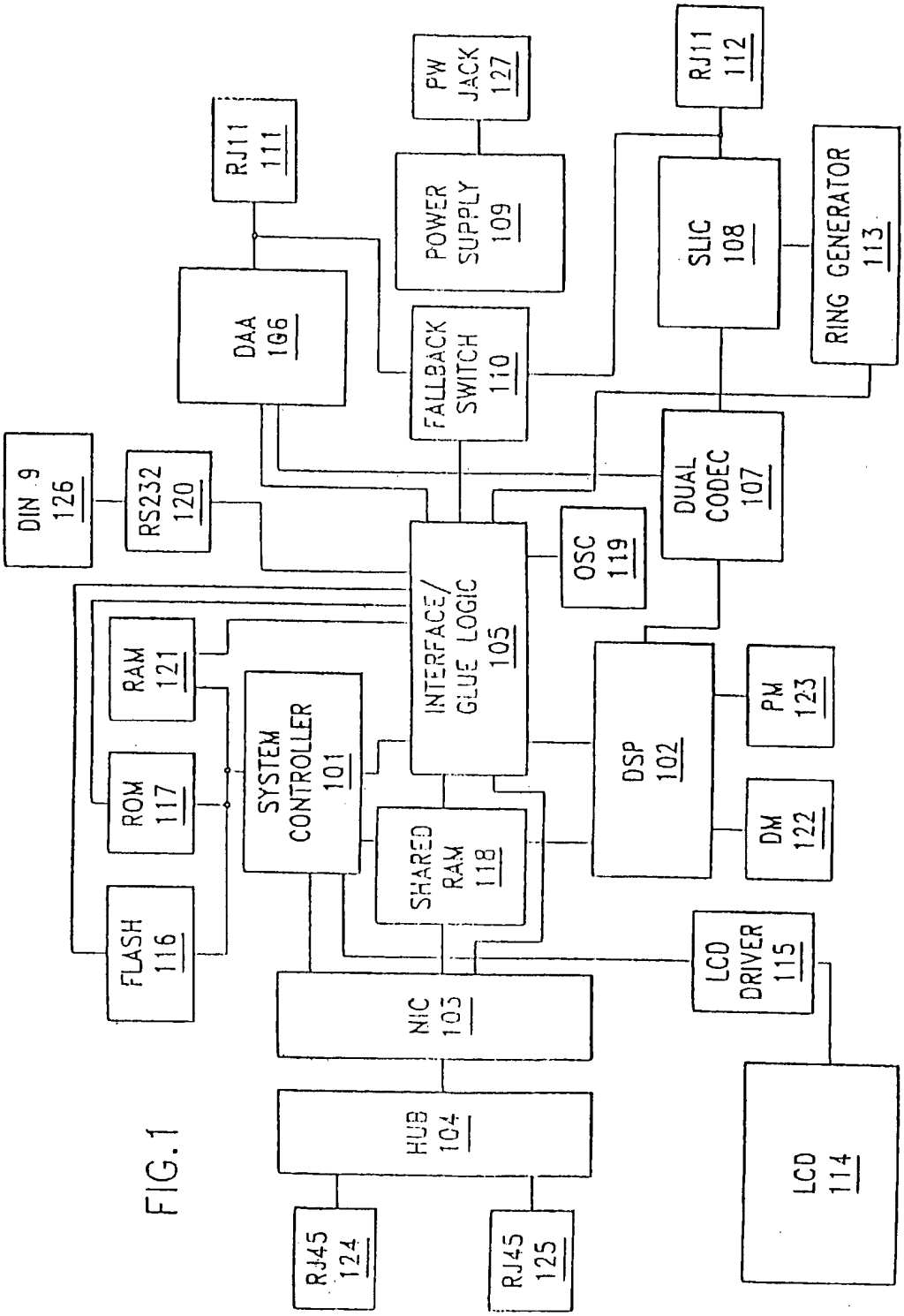

RJ11 111
PW JACK 127
RJ11 112
POWER SUPPLY 109
SLIC 108
RING GENERATOR 113
DAA 106
FALLBACK SWITCH 110
DIN 9 126
RS232 120
DUAL CODEC 107
OSC 119
INTERFACE/ GLUE LOGIC 105
RAM 121
SYSTEM CONTROLLER 101
DSP 102
PM 123
ROM 117
SHARED RAM 118
DM 122
FLASH 116
LCD DRIVER 115
NIC 103
HUB 104
LCD 114
RJ45 124
RJ45 125

FIG. 1

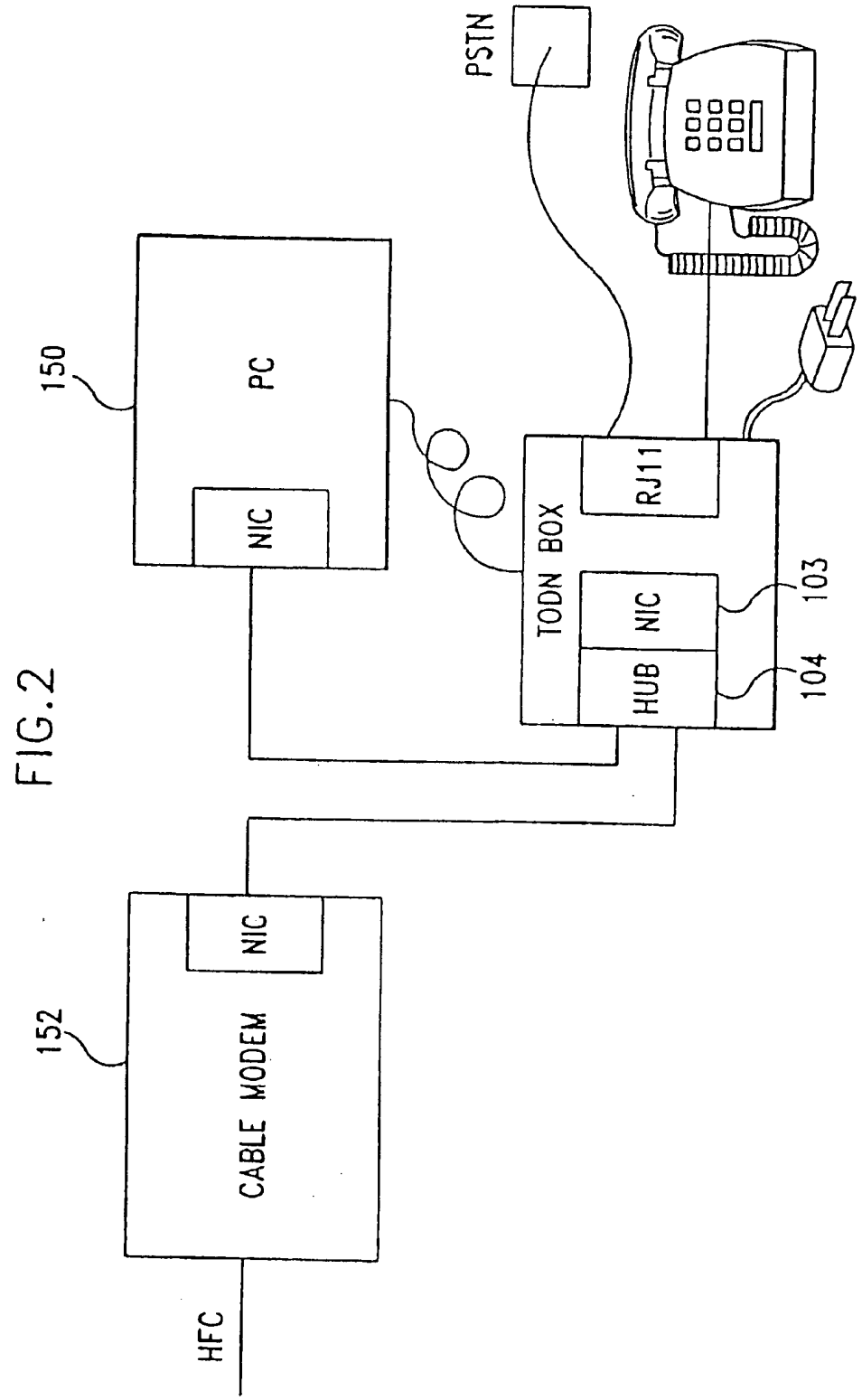

PSTN
150
PC
NIC
TODN BOX
RJ11
NIC
HUB
103
104
152
NIC
CABLE MODEM
HFC

FIG. 2

FIG.3

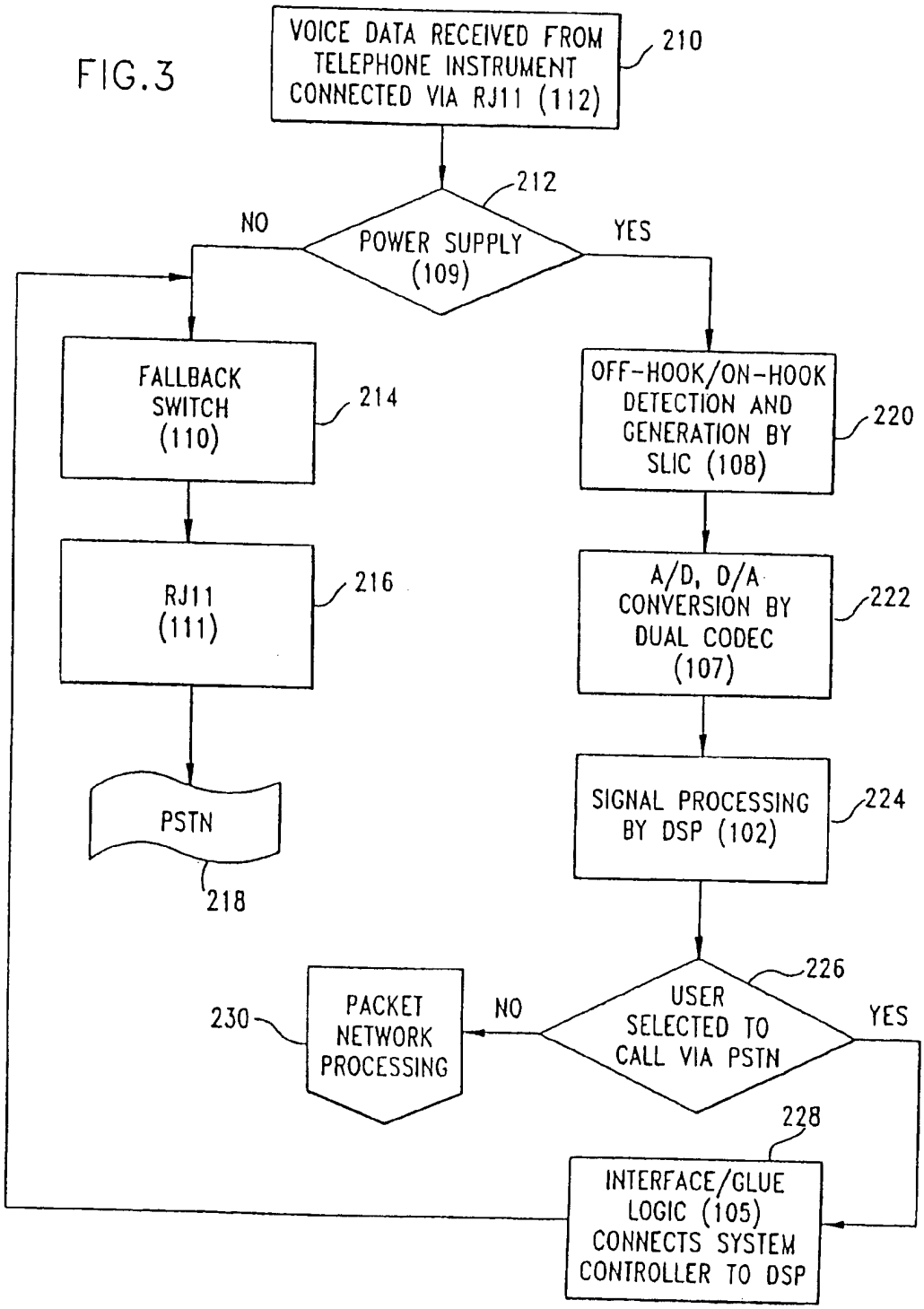

VOICE DATA RECEIVED FROM TELEPHONE INSTRUMENT CONNECTED VIA RJ11 (112)
210
212
POWER SUPPLY (109)
NO
YES
FALLBACK SWITCH (110)
214
RJ11 (111)
216
PSTN
218
OFF-HOOK/ON-HOOK DETECTION AND GENERATION BY SLIC (108)
220
A/D, D/A CONVERSION BY DUAL CODEC (107)
222
SIGNAL PROCESSING BY DSP (102)
224
226
USER SELECTED TO CALL VIA PSTN
NO
YES
230
PACKET NETWORK PROCESSING
228
INTERFACE/GLUE LOGIC (105) CONNECTS SYSTEM CONTROLLER TO DSP

FIG.4

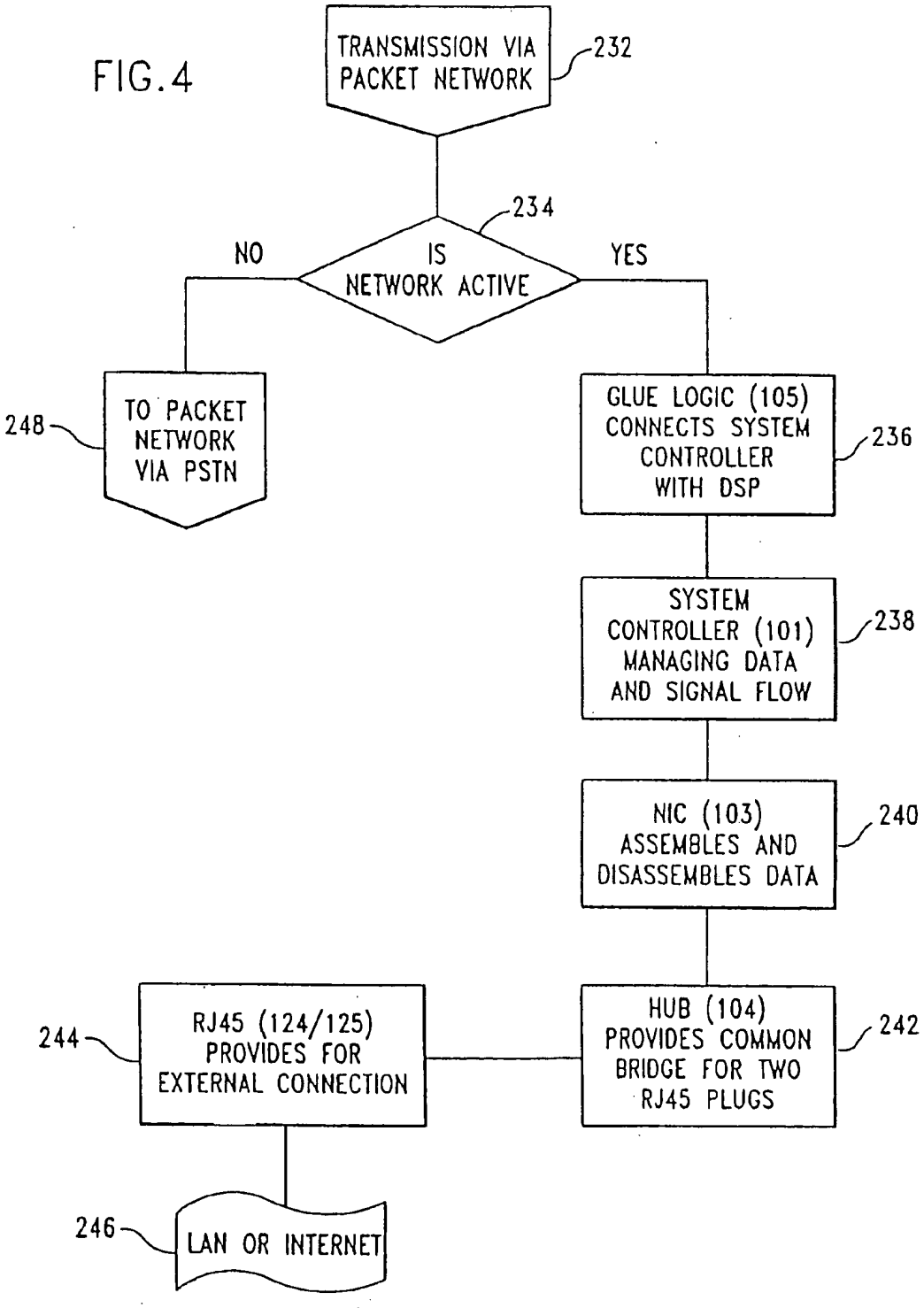

TRANSMISSION VIA PACKET NETWORK
232
234
IS NETWORK ACTIVE
NO
YES
248
TO PACKET NETWORK VIA PSTN
GLUE LOGIC (105) CONNECTS SYSTEM CONTROLLER WITH DSP
236
SYSTEM CONTROLLER (101) MANAGING DATA AND SIGNAL FLOW
238
NIC (103) ASSEMBLES AND DISASSEMBLES DATA
240
244
RJ45 (124/125) PROVIDES FOR EXTERNAL CONNECTION
HUB (104) PROVIDES COMMON BRIDGE FOR TWO RJ45 PLUGS
242
246
LAN OR INTERNET

FIG.5

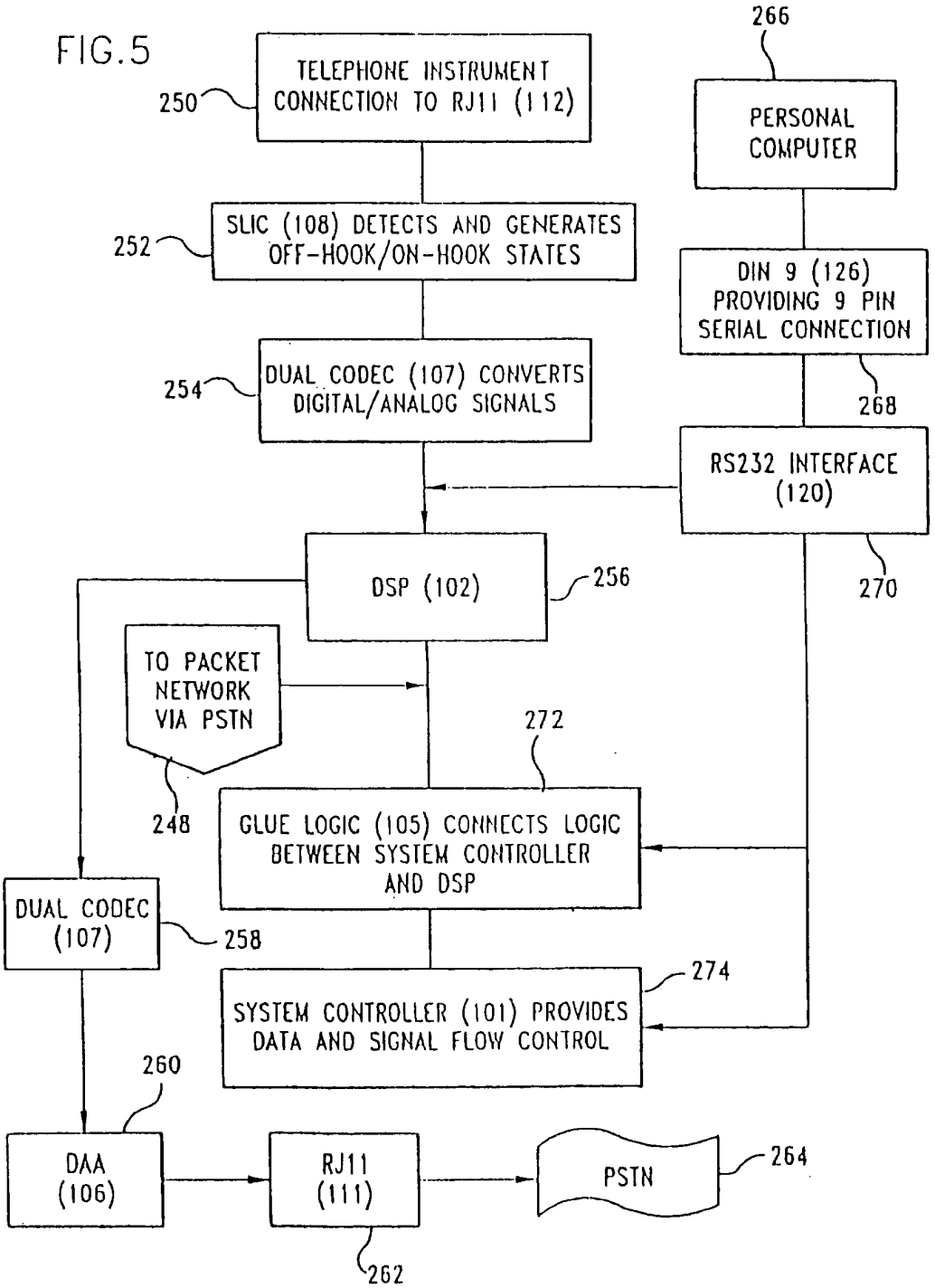

TELEPHONE INSTRUMENT CONNECTION TO RJ11 (112)
250
SLIC (108) DETECTS AND GENERATES OFF-HOOK/ON-HOOK STATES
252
DUAL CODEC (107) CONVERTS DIGITAL/ANALOG SIGNALS
254
DSP (102)
256
TO PACKET NETWORK VIA PSTN
248
GLUE LOGIC (105) CONNECTS LOGIC BETWEEN SYSTEM CONTROLLER AND DSP
272
DUAL CODEC (107)
258
SYSTEM CONTROLLER (101) PROVIDES DATA AND SIGNAL FLOW CONTROL
274
DAA (106)
260
RJ11 (111)
262
PSTN
264
PERSONAL COMPUTER
266
DIN 9 (126) PROVIDING 9 PIN SERIAL CONNECTION
268
RS232 INTERFACE (120)
270

FIG.6

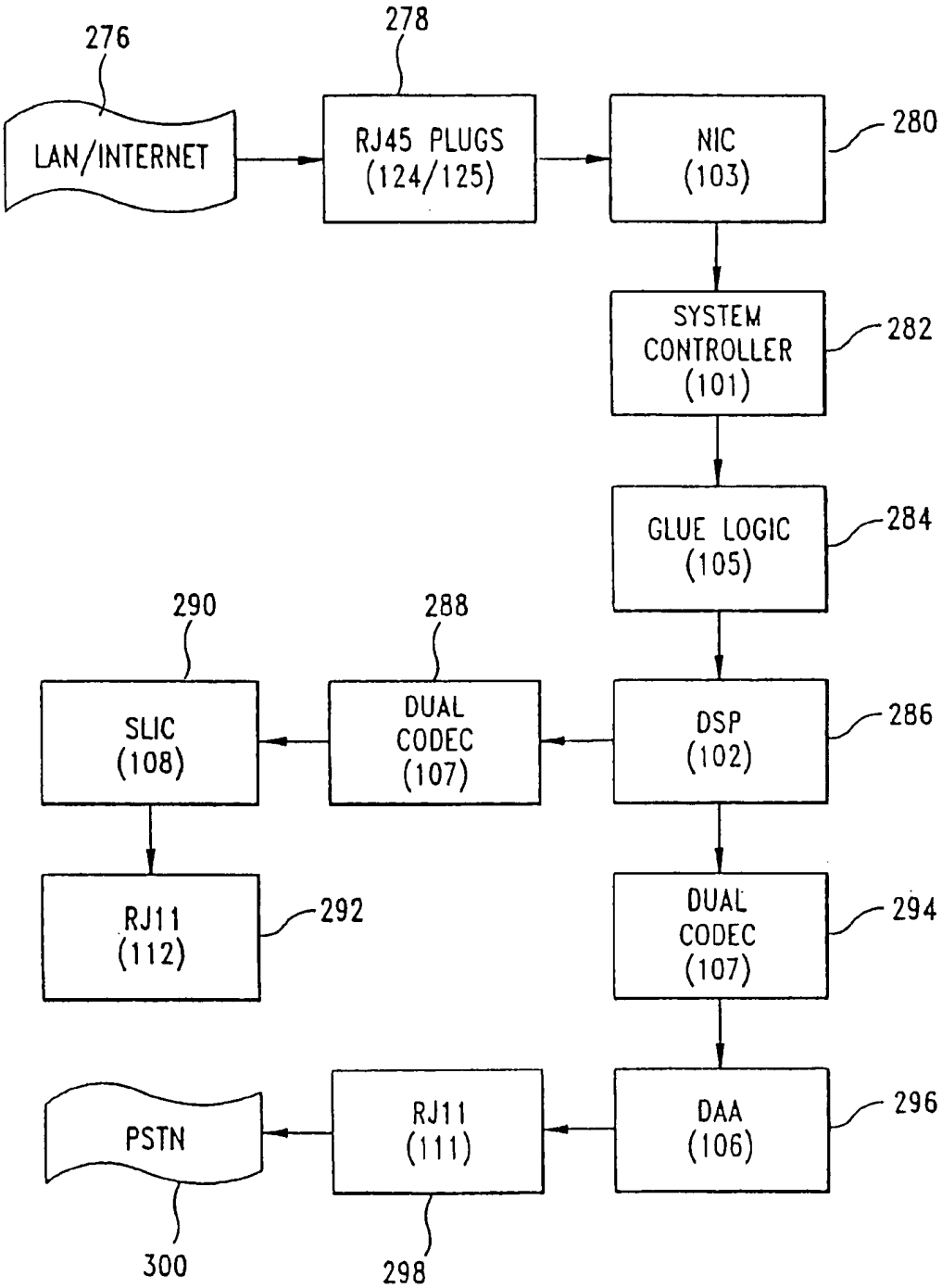
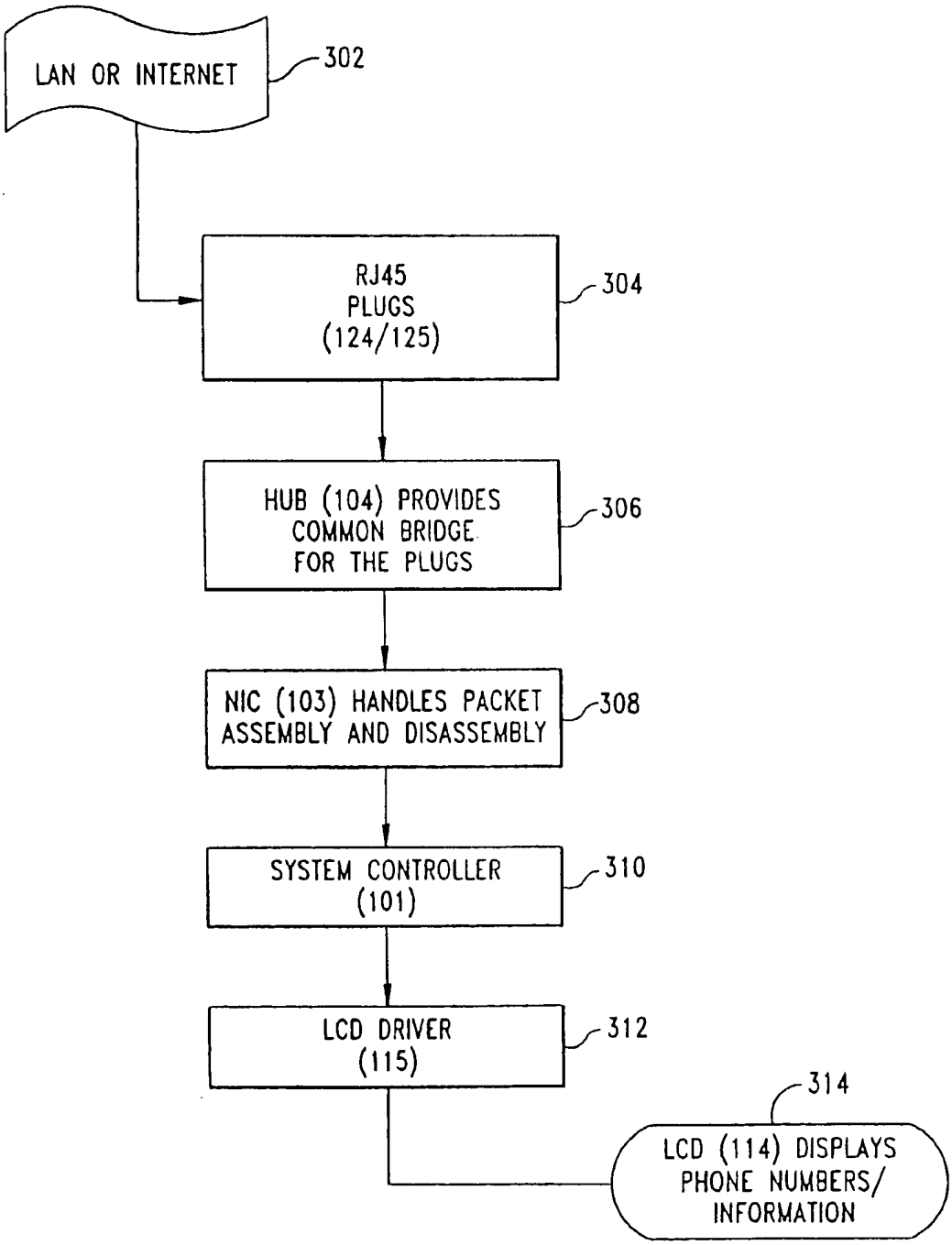

276
LAN/INTERNET
278
RJ45 PLUGS
(124/125)
NIC
(103)
280
SYSTEM
CONTROLLER
(101)
282
GLUE LOGIC
(105)
284
290
SLIC
(108)
288
DUAL
CODEC
(107)
DSP
(102)
286
RJ11
(112)
292
DUAL
CODEC
(107)
294
PSTN
RJ11
(111)
DAA
(106)
296
300
298

FIG. 7


LAN OR INTERNET
302
RJ45 PLUGS (124/125)
304
HUB (104) PROVIDES COMMON BRIDGE FOR THE PLUGS
306
NIC (103) HANDLES PACKET ASSEMBLY AND DISASSEMBLY
308
SYSTEM CONTROLLER (101)
310
LCD DRIVER (115)
312
314
LCD (114) DISPLAYS PHONE NUMBERS/ INFORMATION

PACKET NETWORK TELEPHONE INTERFACE SYSTEM FOR POTS

TECHNICAL FIELD

This invention relates generally to communication systems, and more particularly, to a device for communicating voice data over a packet-based data network.

BACKGROUND ART

Currently, programs are available for digitizing the voice data as a person talks, and sending the digital data over the Internet. The existing programs are designed to run on a personal computer having a sound card with speakers and microphones. Accordingly, the only way to utilize the existing programs is to power on the personal computer before getting a voice connection through the Internet. In addition, the programs are capable of being used only in conjunction with a browser, for example, when navigating on the World Wide Web (Web) over the Internet. Moreover, the existing programs allow calling only those on the Internet having a compatible phone software, that is, a same type of compression scheme. They also do not automatically provide a capability for calling a party on the public switched telephone network, unless another connection is made to a "phone gateway".

The U.S. Pat. No. 5,526,353 by Henley et al. is a system and method for communicating audio data in a packet-based computer network where transmission of data packets through the computer network requires variable periods of transmission time. The system described in this patent requires a PC connection for delivering voice data over the packet network. It also does not include any other additional functions the present invention is able to provide.

SUMMARY OF THE INVENTION

The present invention is a standalone packet network phone interface unit which enables a plain old telephone instrument or a fax machine to be connected directly to a packet network. The present invention may be used without having a personal computer (PC), and provides, inter alia, features such as a plain old telephone set (POTS) interface to a packet network, a "phone gateway" capability, and an "IP phone" capability. Yet another feature of the present invention is an ability to connect to various serial communications mediums such as the cable TV channels and the integrated services digital network (ISDN). Such connections are useful for making telephone calls that bypass a local loop and/or local telephone company. Other features provided by the present invention include an ability to identify a caller id and display the information; a switch option to connect directly to a public switched telephone network (PSTN) line; and, an interface to connect to a data terminal equipment such as a PC.

The apparatus of the invention includes an analog to digital/digital to analog converter for converting voice signals to digital signals and vice versa; a signal processor for filtering and encoding the converted digital signals whose functions may also include operating as a modem; a network interface card for providing a physical connection to a network for transmitting and receiving packets of data from a network; a system controller for controlling and manipulating data, signals and their flow; and various additional interfacing devices for connecting to the external mediums, including a two-wire to four-wire converter for electrically

connecting the telephone instrument. The system of this invention may further include an erasable and electrically programmable read only memory, said memory having capacity to be programmed remotely and dynamically.

The apparatus of the present invention is connected to a packet network and receives and transmits real time data such as voice samples from and to the network. The apparatus includes a network interface card and a controller for receiving packets from the network having an IP address corresponding to the network address of the apparatus. The controller strips off headers and trailers from each data packet received. The stripped data packets are generated into a stream of digital data which then is converted to analog audio data and transmitted to a telephone instrument. Similarly, audio data received from the telephone instrument is converted and compressed into digital data, assembled into packets with headers and trailers, and transmitted to the network via the network interface card. Accordingly, it is the object of this invention to provide a standalone interface between the POTS and the packet-based data network such that the apparatus of the present invention may be used as a terminal equipment of a packet data network phone system.

The present invention also functions as a one line PSTN gateway to the packet network by utilizing the data access arrangement (DAA). The DAA is directly connected to an analog line and interfaces with the PSTN. The DAA module performs, inter alia, over voltage protection, two-wire to four-wire signal conversion, ring detection, and off-hook on-hook detection and generation.

When the present system is functioning as a "phone gateway", a signal processor incorporated in the present invention performs dial tone detection and generation, dual tone multi-frequency (DTMF) tone generation and detection, voice compression and decompression, transmissions of off-hook signals to the DAA, and echo cancellation and suppression.

The system controller, also incorporated in the present invention, plays a pre-recorded message to announce to a caller on the PSTN side to press the destination number. The system controller then transfers the number to the telephony server on the digital data network to map it to an appropriate network address such as an IP address. The telephony server returns the network address to the system controller in the present invention to be used for further communication. If the call is initiated from the digital data network side, the system controller takes the destination number and passes it to the signal controller which then generates DTMF tones to make a call out to the PSTN. Accordingly, it is yet another object of this invention to provide a public switched telephone network gateway to a digital network phone system.

The apparatus of the present invention also provides means for connecting a data terminal equipment such as a PC. With a PC connection to the system of the present invention, many different feats may be accomplished. For example, a user may download via the PC a program for simultaneous voice data (SVD) function. The SVD function program enables the user with a single telephone line to speak on a telephone via Internet telephony while at the same time navigating among Web pages. In this way, the PC need not be equipped with a sound card, a microphone, or a speaker when making a phone call to the IP network. Yet, by downloading the SVD function program, the present system may effectively function as an "IP phone".

Other uses for connecting to a data terminal include information exchange for speed dial, automatic dialer, call detail records, dynamic firmware upgrade, caller identifica-

tion record, voice mail, and normal everyday computer interface. Accordingly, it is a further object of this invention to provide an interface for connecting the present invention with a data terminal equipment such as a PC.

The present invention also includes a fallback switch for connecting a telephone instrument directly to the PSTN. This switch provides an option for connecting either to the PSTN line or a packet network. When the system of the present invention detects a power-off state, it automatically provides a direct connection from the telephone set to the PSTN line. That is, when a power is turned off on the apparatus of the present invention or if a power failure occurs, a phone call will be made directly through a telephone line, not any different from the way a plain everyday telephone call is made.

In addition, the system provides an option for the user to connect directly through the PSTN line instead of communicating through the packet network. That is, the user may choose to use a telephone set connected to the system to make a call via a conventional telephone line and the PSTN. The system includes a logic for determining whether a call will be made via a packet network or the PSTN. This option may be provided to the user by way of a predetermined number. For example, when the user dials #9 as his first digit when making a call, the system determines that a connection is to be made directly through the PSTN line, and activates the fallback switch. Accordingly, the fallback switch provides a direct connection to the PSTN line. Therefore, it is yet another object of this invention to provide an optional arrangement capable of making connection to either the PSTN or the packet network.

The present invention also provides a user with an option to connect to a number of serial communications lines by including a hub. The hub connects a network interface card to another external medium such as a cable TV system or an ISDN. Using a cable may get a user to a long distance carrier, which would bypass both the local loop and local telephone company, thus avoiding many bottleneck problems associated with a local loop. Having an ISDN provides an end-to-end digital connectivity with full-use, clear channels and a standardized interface. Accordingly, it is yet another object of this invention to provide a common bridge to a plurality of serial line communications interfaces.

The present invention includes a memory module capable of storing 60 minutes worth of voice data. Therefore, it is another object of this invention to provide an answering and messaging system for the digital packet data network phone system.

BRIEF DESCRIPTION OF THE DRAWINGS

FIG. **1** is a block diagram illustrating internal interconnections, including the present invention.

FIG. **2** is an exemplary illustration showing connections to a packet network via the three-way internal hub.

FIG. **3** is a flow diagram illustrating a possible path for data and signal flow in the components of FIG. **1** when power supply is down.

FIG. **4** is a flow diagram illustrating a possible path for data and signal flow in the components of FIG. **1** for communicating audio data in a packet-based network when the unit in FIG. **1** is connected via a LAN/Internet.

FIG. **5** is a flow diagram illustrating a possible path for data and signal flow in the components of FIG. **1** for communicating audio data in a packet-based network when the unit in FIG. **1** is connected via the PSTN.

FIG. **6** is a flow diagram illustrating a possible path for data and signal flow in the components of FIG. **1** when the unit in FIG. **1** is functioning as a medium for interconnecting packet data network and the PSTN, that is, a "phone gateway".

FIG. **7** is a flow diagram illustrating a possible path for data and signal flow in the components of FIG. **1** for displaying information such as a caller id.

DESCRIPTION OF THE PREFERRED EMBODIMENT

FIG. **1** is a block diagram illustrating an exemplary layout of the internal components of the present invention, a packet network telephone interface system for POTS, in its preferred embodiment. The main components shown in FIG. **1** are: a system controller **101**; a digital signal processor (DSP) **102**; a network interface circuit (NIC) **103**; a data access arrangement (DAA) **106**; a subscriber line interface circuit (SLIC) **108**; a dual code/decoder (dual CODEC) **107**, one coupled to the SLIC (**108**) and another coupled to the (DAA) **106**; an LCD display **114**; a flash memory **116**; a read only memory (ROM) **117**; a random access memory (RAM) **121**; shared memory **118**; DSP data memory (DM) **122**; DSP program memory (PM) **123**; a fallback switch **110**; a three way hub **104**; an oscillator **119**; power supply circuits **109**; an RS232 driver **120**; an interface/glue logics **105**; and various connectors **111, 112, 124, 125, 126, 127**. Although a single powerful processor can be adapted instead of having a separate system controller and digital signal processor, the preferred embodiment in FIG. **1** is shown as implemented using two separate processors. Such a separation clearly defines functional boundaries and enhances determination and easy isolation of potential problems.

Typically, a telephone instrument is connected to the SLIC **108** via the RJ11 **112**. The SLIC **108** receives a voice signal from the telephone instrument and performs functions such as detecting and generating off hook and on hook states. The SLIC **108** is coupled to the dual CODEC **107**. The dual CODEC **107** converts the voice signal into digital data. The digital signal processor (DSP) **102** generally filters, encodes, compresses/decompresses, and performs echo cancellation and suppression on the data. Additional functions of the DSP **102** include dial tone detection and generation, DTMF detection and generation, transmitting off-hook signals to the DAA, and operating as a modem. The data is also made available to the system controller **101** via the interface/glue logic **105**. The system controller **101** and the network interface card (NIC) **103** together filters and packages data into data packets, adds headers and trailers, including error check bits, for the appropriate communications protocols, for example, TCP/IP. The digitized voice data is then transmitted to the packet network via the NIC **103** and the connection at RJ45 plugs **124, 125**.

Similarly, when a packet data is received by the present invention via the connections at RJ45 plugs **124, 125** and the NIC **103**, the system controller **101** and NIC **103** together filters out data packets received, and strips off headers and trailers while checking the CRC error. The DSP **102** also processes the data, decompressing the data when voice compression is detected. The dual CODEC **107** converts the processed digital signals into voice signals for transmission to a telephone instrument connected to the SLIC **108** via the RJ11 **112**. At the same time the DSP **102** drives the SLIC **108** by performing a call process tone generation.

The system controller **101** also controls the host interface **126**, LCD display **114**, updating the FLASH memory **116**, as

well as the state machines for the telephone states. The minimum set of telephone states include idle, dial tone, DTMF detection, busy tone, ring back tone, voice, and ring. The system controller **101** switches the states based on telephone events such as off-hook detection, on-hook detection, DTMF tone detection, busy detection, not busy detection, circuit connected, call terminated, ring start, and ring stop.

The system controller **101** is generally responsible for handling all communication protocol stacks for adhering to the protocol, for example, TCP-UDP/IP protocols, while managing the data flow. The system controller **101** together with NIC **103** handles all the layers of the communication protocols including point to point protocols (PPP) or serial line internet protocol (SLIP) and obtains the internet protocol (IP) address if a permanent address is not available. For example, if a permanent network address for the unit of the present invention is not assigned, the system controller **101** acquires a temporary address from an address pool by adhering to protocols as PPP. Once the network address is acquired, NIC **103** exchanges its application addresses, also known as "port address", with the server if a permanent port address is not assigned. The combined address of the network and the port addresses together is often called socket address. The NIC **103** receives only the packet that contains its network address and application address when it detects the incoming real time data from the network.

The telephone signaling message preferably uses a distinctive application address from the voice samples so that the NIC **103** can easily distinguish the voice from the signal. In the preferred embodiment, guaranteed protocols such as TCP are used to deliver the signaling, while datagram such as UDP may be adapted for voice samples.

The configuration of the data format may be exchanged through a proper handshake protocols such as H.323 over the network. A fixed format can also be utilized by configuring the format via the PC interface. In the preferred embodiment, the information about the duration of the packet and the type of compression scheme used is exchanged via the proprietary protocols over the network. The default setting is set initially at the manufacturing time and may be modified via the PC interface over an RS232 serial link **120**.

In the preferred embodiment of the present invention, the system controller **101** reads instructions stored in a ROM **117**. These instructions include dynamically updating programs and data in the FLASH memory **116**. The system controller **101** for its subsequent processing accesses the updated programs and data in the FLASH memory **116**. In addition, the updated programs and data are downloaded to the DSP program memory (PM) **123** and data memory (DM) **122** when the system is powered on. The DSP **102** reads PM **123** and DM **122** for its processing instructions and data access.

The NIC **103** is connected to an interface/glue logic **105** that supports direct memory access to the shared memory **118**. In the preferred embodiment, an Ethernet 10baseT NIC **103** is used, however, the disclosure does not limit to the Ethernet only. A token ring or an asynchronous transport module (ATM) network interface circuit and associated controllers may be substituted. A three way hub **104** is added to the unit in order to reduce the users' burden of requiring an external hub in case a link to an external communications line is needed. By having one end of the three way 10baseT Ethernet hub **104** connected to a 10baseT Ethernet NIC **103**, it is possible to have a three way connection, for example, among a cable modem via serial connection **124/125**, a personal computer and the present invention.

The communications link between a data terminal, for example, a personal computer, and the apparatus of the present invention is shown in FIG. 1 as linked via a RS232 serial link **120**. However, other links are also possible. Depending on the throughput required on the link, any other serial link such as a universal serial link, an IEEE 1394 link, or a parallel links such as IEEE P1284 may be adapted.

The interface/glue logic **105** is designed to assist the NIC **103** to transfer data from and to the shared memory **118** and also has Universal Asynchronous Receive and Transmit (UART) compatible serial interface logic for the host interface **126**. Generally, the system controller **101** accesses the interface/glue logic **105**. The interface/glue logic **105** also includes message interchange logic to exchange messages between the DSP **102** and the system controller **101**. Also included in the interface/glue logic **105** is a clock generator which takes a clock from a oscillator **119** as a master clock and divides it down to various rates required by each components in the unit. The system interface/glue logic **105** includes a logical connection between the system controller **101** and the LCD driver **115** such that the system controller **101** may display short messages on a strip of LCD **114**. The messages displayed on the LCD **114** in the preferred embodiment are caller identification and the system status.

The shared memory **118** may be used as a message passing medium between the system controller **101** and the DSP **102**. For example, when data is received through the network interface card, the system controller **101** handles the received data first, then the system controller **101** writes the received data onto the shared memory **118**. The DSP **102** can then directly read the shared memory **118** to retrieve the system controller **101** processed data. In addition, the system controller **101** may access the data ready for transmission, written on the shared memory **118** by the DSP **102**, and attach TCP-UDP/IP headers and trailers before transmitting to the network.

The functions of DSP **102** include compressing and decompressing voice data, DTMF detection, echo cancellation as well as call process tone generation when driving the SLIC **108**. The DSP is also used to implement the data pump for a data modem when the network is connected through an analog phone line. The unit may be connected to an analog line on the PSTN via the DAA module **106**, which is connected to a dual CODEC **107**. The DAA module **106** handles any over voltage protection, 2-wire to 4-wire conversion, ring detection, and on-hook and off-hook detection and generation.

The two wire on the RJ11 **111** connector on the DAA side are connected to the corresponding two wires on the RJ11 **112** connector on the SLIC **108** side via a fallback switch **110**. When a power is shutdown on the unit or a power failure occurs, the fallback switch **110** connects the two RJ11 **111**, **112** connectors. The fallback switch **110** also connects the two RJ11 **111**, **112** connectors upon a software command from the system controller. Thus, the fallback switch **110** may be used to connect the telephone instrument on the SLIC **112** side to the PSTN in case of a power failure or a packet network failure. In addition, the fallback switch **110** supports optional arrangements for connecting either to the PSTN or the packet network even when the power supply is available.

In the preferred embodiment, the present invention may also be used as a one line digital packet data network to PSTN gateway. In this mode, the DSP's **102** functions

include voice compression/decompression, echo cancellation, DTMF tone generation and DTMF tone detection. The system controller **101** plays pre-recorded message to announce to a caller on the PSTN side to press the destination number, then transfers the number to the telephony server on the digital packet data network. The telephony server then maps the number to an appropriate IP address and returns the IP address to the apparatus of the present invention. The apparatus of the present invention then uses the IP address for further communication. When the call is initiated from the digital packet data network side, the system controller **101** takes the destination number and passes it onto the DSP **102** so that the DSP **102** can call out to the PSTN by generating the DTMF tones.

The unit contains a large enough memory to store at least 60 minutes of compressed voice so that it can be used as an answering machine.

The dual CODEC **107** is also connected to a subscriber line interface circuit (SLIC) **108** to convert 2 wire telephone signals to 4 wire signals. The SLIC **108** is connected to a ring generator **113** for generating a telephone ring on the instrument.

The apparatus of the present invention is powered via a power jack **127** by a power supply that converts the household AC power to the required DC voltage levels. When the AC power is not readily available, the unit may also be powered by a power supply that converts backup battery DC power to the required DC voltage.

The unit of the present invention may be connected locally or remotely to a data network in a similar manner as any other data terminal equipment such as a PC. overview configuration of the network component interconnections, including the present invention. In For example a workgroup may include a plurality of nodes all connected to an external hub which is used to link together all the nodes in a workgroup. The plurality of nodes may further include the units of the present invention with telephone instrument connections. In addition, the unit may be linked to the PSTN via a line connecting RJ11 plug on the unit **111** and a socket in a wall for physical telephone line connection. The hub may be linked to a router which provides a bridge connection between a workgroup and an outside network. Through this configuration, the data may travel from and to the various networks. For example, a packet data originating at some point in the network travels via the router, the hub, the unit of the present invention, to the PSTN, terminating in a node located somewhere in the PSTN. Similarly, a packet data originating in the network follows the same path and terminates at the telephone instrument connected at the unit of the present invention. Reverse path for both situations are, of course, possible. A direct link from the unit to a PC via connections such as RS232 interface is also made possible by the present invention.

FIG. **2** is an exemplary illustration showing connections to a packet network via the three-way internal hub. The hub **104** connects the NIC **103** in the unit with a PC **150** and also a cable modem **152** via two RJ45 plugs. With the illustrated connection, a telephone voice communication via cable line is made possible as explained previously in reference to the FIG. **1**. The cable modem **152** in FIG. **2** is shown only as an example and not as a limitation. Any other network connecting medium capable of carrying packet data may be linked to the hub **104**.

FIG. **3** is a flow diagram illustrating a possible path for data and signal flow in the circuit components of FIG. **1** when a power supply is down. In step **210**, voice data is received into the unit via a telephone instrument connected to the unit at RJ11 plug **112**. In step **212**, a test is made to determine whether a power supply is available. In step **214**, when no power is supplied to the unit, a fallback switch **110** automatically links a line from the RJ11 plug **112** to the other RJ11 plug **111**. In step **216**, the RJ11 plug **111** is physically connected to the PSTN, enabling voice data to be communicated via the PSTN **218**.

FIG. **3** also illustrates a logical data and signal path in the circuit components in FIG. **1** when a power supply is available, but a user optionally elects to link the telephone instrument directly to the PSTN. In step **220**, SLIC **108** whose functions include detecting and generating off-hook and on-hook telephone states, detects off-hook condition. In step **222**, the dual CODEC **107**, utilized for performing analog to digital and digital to analog signal conversions, generates digital signals from the audio data received via the telephone instrument at step **210**. In step **224**, the DSP **102** whose functions include dial tone detection, DTMF detection and generation, voice processing including voice compression and decompression, modem processing, and transmitting off-hook signals to DAA, processes the converted digital signals accordingly. In step **226**, the DSP **102** determines whether an option was selected for a direct connection to the PSTN. In step **228**, the DSP **102** informs the system controller **101** via the interface/glue logic **105** the selected option to establish a direct line to the PSTN. The system controller **101**, again via the interface/glue logic **105**, activates the fallback switch which establishes a direct connection as explained previously.

In step **226**, if at DSP **102** it is determined that the audio data is to be communicated via packet network, a normal proceeding continues in step **230**, as will be explained in reference to FIG. **4** and FIG. **5**.

FIG. **4** is a flow diagram illustrating a possible path for data and signal flow in the components of FIG. **1** for communicating audio data in a packet-based network when the unit in FIG. **1** is connected via a LAN/Internet. At **232**, transmission via a packet network continues from **230** in FIG. **3**. In step **234**, a query is made to determine whether an active network connection via the NIC **103** exists. When network is not active, communication to the network will be made via the PSTN as shown in step **248** and processing will continue as will be explained in reference to FIG. **5**. If a network connection exists, the glue logic **105**, in step **236**, establishes itself as an interfacing medium for connecting logics between the system controller **101** and the DSP **102**. In step **238**, the system controller **101** whose functions include handling network protocol operations, controlling data and signal flows, correcting error, and managing simultaneous voice and data flow, prepares the data accordingly. In step **240**, the NIC **103** assembles the data into packets, readying the packets for transmission via the NIC **103**. In step **242**, the hub **104** provides a common bridge among the NIC **103** and the two RJ45 plugs at **124** and **125** in FIG. **1**. The two plugs in step **244** then provide outlets to external communications lines, whereby the packetized data travels to its destination node via the packet network.

The data and signal flow described above refers to one direction, namely, voice data to packet network. Of course, the packet data received into the system would follow the equivalent logic flow but in reverse order whereby packets are received at either of the two RJ45 plugs **124**, **125** in FIG. **1**, disassembled and converted to analog signals to be transmitted to their destination nodes via the two RJ11 plugs at **111** and **112** in FIG. **1**.

FIG. **5** is a flow diagram illustrating a possible path for data and signal flow in the circuit components of FIG. **1** for

communicating audio data in a packet-based network when the unit in FIG. **1** is connected via the PSTN. In step **250**, voice data is received into the unit via a telephone instrument connected to the unit at RJ11 plug **112**. In step **252**, SLIC **108** whose functions include detecting off-hook state, processes accordingly. In step **254**, the dual CODEC **107**, converts the audio data received via the telephone instrument at step **210** to digital signal. In step **256**, the DSP **102**, in addition to performing its other functions, operates as a modem. In its capacity as a modem, the DSP modulates and demodulates the filtered digital signal, transmits off-hook signal to the DAA for communications via the PSTN. In step **258**, the dual CODEC **107** converts the digital signal back to analog signals. In step **260**, the DAA **106**, a telephone-system protective device, is used to interface to the telephone network shown in step **264**. The telephone network is connected through the RJ11 plug **111** in step **262**.

FIG. **5** also illustrates an example logic flow among the components of the unit in FIG. **1** when a PC is connected to the unit via DIN9 connection and RS232 interface as shown in steps **266**, **268**, **270**. The PC connected via RS232 interface interacts with the system controller **101** and the DSP **102** through a connection to the glue logic **272** as shown in step **272**, **256**, **274**. The DSP **102** may again acts as a modem between the PC and the PSTN.

The processing performed in steps **272**, **274**, **256**, **258**, **260**, **262**, and **264** as described previously also follows the step **248** in FIG. **4**, when the network connection via LAN/Internet is not available.

FIG. **6** is a flow diagram illustrating a possible path for data and signal flow in the components of FIG. **1** when the unit in FIG. **1** is functioning as a medium for interconnecting packet data network and the PSTN, for example, a "phone gateway". In FIG. **6**, the packets of data are received into the NIC **103** in step **280** via the RJ45 plugs **124**, **125** in step **278** from the LAN/Internet network in step **276**. In step **280**, NIC **103** disassembles the packets into digital signals while in step **282**, the system controller **101** handles protocol formatting and error checking. In step **284**, the glue logic functions as a logical connector between the system controller **101** and the DSP **102**. In step **286**, the DSP process the digital signals, performs dial tone detection, DTMF generation and detection, and voice compression and decompression when detected. In step **288**, the dual CODEC **107**, one of which is coupled with the SLIC **108** converts digital signals into analog signals. In step **290**, the SLIC **108** whose function includes on-hook, off-hook detection and generation, detects an appropriate condition and triggers a ring generator **113** in FIG. **1** when the telephone instrument connected via the RJ11 **112** in step **292** is available for establishing a conversation. Any subsequent exchange of data from the telephone instrument connected at RJ11 **112** and the LAN/Internet network takes place as described.

Alternatively, the digital signal processed data may be processed as shown in step **294** through the dual CODEC, one of which is coupled to the DAA **106**. The DAA **106**, in step **296**, interfaces on-hook, off-hook states with a telephone switching system via the RJ11 plug **111** in step **298** for communications over the PSTN in step **300**. The processing described effectively allows the unit of the present invention to function as a "phone gateway" between the packet network and the PSTN.

FIG. **7** is a flow diagram illustrating a possible path for data and signal flow in the components of FIG. **1** when displaying information such as a caller id. In steps **302** and **304**, packet data enters the unit via the RJ45 plugs **124**, **125**. Step **306** shows how the hub **104** may function as a common bridge connecting the two RJ45 plugs **124**, **125** with the NIC **103**. The NIC **103** in step **308** then disassembles the packet data while the system controller in step **310** performs error checking and strips off appropriate protocol header and trailers. In addition, the system controller **101** performs information mapping with received phone number. In step **312**, the information and phone number are relayed to the LCD driver **115** which displays the information on the LCD **114** in step **314**.

While the invention has been particularly shown and described with respect to a preferred embodiment thereof, it will be understood by those skilled in the art that the foregoing and other changes in form and details may be made therein without departing from the spirit and scope of the invention.

Having thus described our invention, what we claim as new, and desire to secure by Letters Patent is:

1. A packet network telephone interface system for enabling communications with plain old telephone sets over a plurality of telecommunications networks, said system comprising:

telephone input/output units for receiving signals from and transmitting signals to a plain old telephone set and a public switched telephone network;

a network connection input/output unit for receiving data packets directly from and transmitting data packets directly to a data packet based network having a defined network protocol;

a processing subsystem including a network interface card, located in series between the network connection unit and the telephone input/output units, for converting signals from the plain old telephone set into data packets in accordance with said defined network protocol for transmission over the data packet based network, and for converting data packets from the data packets based network into signals suitable for transmission to the plain old telephone set, the processing subsystem including;

at least one dual coder/decoder connected to the input/output units for converting analog signals into digital signals, and for converting digital signals into analog signals;

switching means to determine whether signals from the plain old telephone are transmitted to the telephone network, or to the network interface card for conversion into data packets for transmission to the data packet based network.

2. The system as in claim **1**, wherein said processing subsystem comprises:

a digital signal processor coupled to said at least one coder/decoder.

3. The system as in claim **1**, wherein said system further includes a ring generator for generating ringing signals; and

said at least one input/output unit includes at least one receptacle plug for receiving connection from a telephone instrument, said receptacle plug further coupled to a subscriber line interface card, whereby said subscriber line interface card detects and generates off hook and on hook states of said telephone instrument, said subscriber line interface card further activating said ring generator for generating ringing signals for said telephone instrument.

4. The system as in claim **1**, wherein said system triggers automatic connection to the public switched telephone network when said system detects a group of predetermined conditions.

5. The system as in claim 1, wherein one of said plurality of interface devices includes a data access arrangement for interfacing to the public switched telephone network and one of said at least one external medium is said public switched telephone network, whereby a telephone call can be made out to said public switched telephone network from said system via said data access arrangement.

6. The system as in claim 5, wherein said system further includes means for enabling said telephone call to be made via said data access arrangement to said public switched telephone network when said system receives data packets via said at least one network interface unit.

7. The system as in claim 1, wherein one of said plurality of interface devices includes a device for connecting a data terminal equipment to said system and said at least one external medium is a data terminal equipment.

8. The system as in claim 7, wherein said data terminal equipment includes a personal computer, far downloading programs to said system from said personal computer.

9. The system as in claim 1, wherein one of said plurality of interface devices includes an LCD driver and said at least one external medium is an LCD, whereby said system may display a plurality of information, which plurality of information includes a caller identification number.

10. The system as in claim 1, wherein one of said plurality of interface devices includes a hub coupled to said network interface unit for connecting said system to a plurality of external communications lines.

11. The system as in claim 10, wherein said hub enables said system to be connected to a packet network.

12. The system as in claim 1, wherein said system further includes a memory device and means for storing voice data messages in said memory device, said means for storing also enabling a party to retrieve said voice data messages, whereby said system can function as a telephone answering/messaging system.

13. The system as in claim 1, further including an erasable and electrically programmable read only memory, said memory having capacity to be programmed remotely and dynamically.

14. An interface system according to claim 1, wherein:

the telephone input/output units include

i) a first input/output unit for connecting the interface system with the plain old telephone set, and

ii) a second input/output unit for connecting the interface system with the public switched telephone network; and

the network connection input/output unit includes a third input/output unit for connecting the interface system with said one of the packet based networks.

15. A method of establishing and maintaining voice communications with POTS over a plurality of communications network services, which plurality includes the PSTN and packet-based networks, said method comprising the steps of:

receiving analog voice signals from a POTS;

establishing a direct link over said PSTN network service when a power supply is not available;

using a telephone interface unit to determine whether to communicate directly over the PSTN or directly over one of the packet-based networks, said one of the networks having a defined network protocol;

preparing said analog voice signals for communications over the determined communication network service including the step of, when preparing said analog signals for communication over said one of the packet-based networks, using a processing subsystem including a network interface card located in the telephone interface unit, to convert the analog signals into data packets in accordance with said defined network protocol.

receiving data from said the determined communication network services;

using a dual coder/decoder, connected to a plurality of input units, a plurality of output units and to the network interface card, to convert streams of digital signals into analog signals and to convert analog signals into digital signals; and

preparing said received data for communications to said POTS.

16. The method according to claim 15, wherein said step of preparing said analog voice signals includes the steps of:

digitizing said analog voice signals;

inserting communications protocol control codes; and

assembling said digitized voice signals into packets,

whereby if said at least one of said plurality of communication network services is packet-based network, said packets are communicated over said packet-based network.

17. The method according to claim 15, wherein said step of preparing said data for communications to said POTS includes the steps of:

disassembling packets of data into stream of data; and

converting said stream of data into analog voice signals,

whereby if said at least one of said plurality of communication network services is packet-based network and said data includes packets of data, said analog voice signals are communicated over said POTS.

18. The method according to claim 15, wherein said step of establishing a direct link over said PSTN network services is performed when a party selects an option to communicate directly to said PSTN network.

19. The method according to claim 15, wherein said step of establishing a direct link over said PSTN network services includes the step of establishing said direct link upon the occurrence of one of a group of predetermined conditions.

20. An interface system according to claim 14, wherein:

the switching means includes a fall-back switch having

i) a first state for connecting together the first and second input/output units for transmitting signals from the plain old telephone set to the public switched telephone network, and

ii) a second state for connecting together the first and third input/output units for transmitting signals from the plain old telephone set to the packet based network; and

the fall back switch automatically changes from the second state to the first state upon the occurrence of one of a set of predefined conditions.

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